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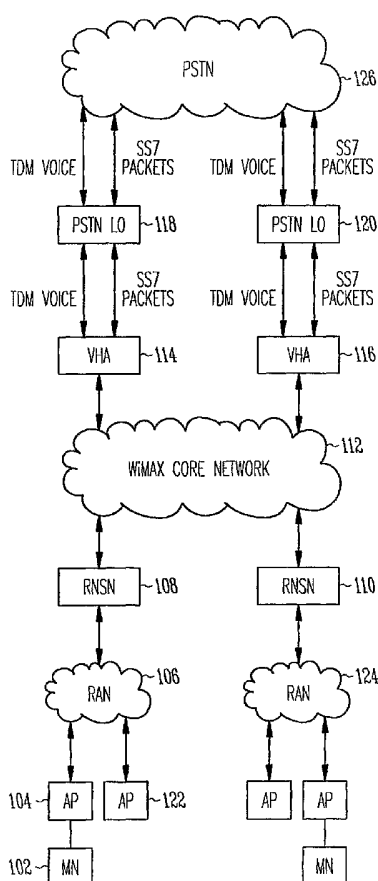
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(54) Title: DISTRIBUTED VOICE NETWORK



(57) Abstract: A method and apparatus (114, 116) that receives an IP packet and encapsulates the packet with an IP header. Further, time-domain multiplexed voice data is received and converted into VoIP packets. Still further, Signaling System (7) (SS7) compliant signals are decoded. The decoded (SS7) signals are received and encapsulated prior to transmission to a telephony device (102).



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DISTRIBUTED VOICE NETWORK

Technical Field

Embodiments of the invention relate to voice-over-IP technology implemented on
5 a mobile wireless broadband network.

Background

Voice-over-IP (VoIP) technology permits parties to communicate orally over a
packet-switched IP network. VoIP technology has grown in popularity, and depending
10 upon certain factors, can offer sound quality that is comparable to that of the public
switched telephone network (PSTN).

Also growing in popularity are wireless mobile networks. Wireless mobile
networks permit a device to link to a network without requiring a physical conductive line
to carry data between the device and the network. Further, such networks permit mobility
15 by allowing a device to change access points in a manner transparent to network elements
or nodes outside of the wireless mobile network domain.

Despite the growing popularity of VoIP technology and wireless mobile networks,
there are no mobile client devices for present VoIP services over the Internet. One factor
that hinders the advancement such mobile devices relates to finding a simple scheme by
20 which a mobile device may be permitted to roam a significant geographic area (and
therefore potentially wander between domains), while appearing keep a single IP address.
The user datagram protocol (UDP) indexes connections by use of a quadruplet that
contains the IP addresses and port number of both connection endpoints. Changing any
one of these four numbers causes the connection to be disrupted and lost. Therefore, it is
25 important that the device appear to keep the same IP address while roaming
geographically. The difficulty in addressing this issue grows as the geographic area
through which a device is permitted to roam grows.

From the foregoing, it is evident that there exists a need for a scheme by which a
wireless IP telephony device can be permitted to roam a geographically significant area,
30 such as a metropolitan area. It is desirable that such a scheme be relatively simple to
implement as an overlay to an existing wireless network. It is further desirable that such a
scheme be easily interconnected to the PSTN.

Brief Description of the Drawings

Figure 1 depicts a network environment in which an embodiment of a voice home agent is deployed.

Figure 2 depicts a protocol stack making up a voice home agent, according to an embodiment of the invention.

Figure 3 depicts a tunneling scheme employed by the mobile IP layer of the protocol stack depicted in Figure 2.

Figure 4 depicts a method of initiating a VoIP phone call, according to an embodiment of the invention.

Figure 5 depicts a method of executing a VoIP phone call, according to an embodiment of the invention.

Figure 6 depicts a hardware environment in which a voice home agent may be embodied, according to an embodiment of the invention.

Detailed Description

Figure 1 depicts a network environment 100 in which one or more mobile nodes 102 may be permitted to roam over a geographically significant area, such as across a metropolitan area. The mobile nodes 102 communicate via digital transmission (typically in the 2-to-6 GHz licensed bands, with typical channel bandwidths ranging from 1.5 to 20 MHz) to an access point 104. An access point (also referred to herein as a base station), such as the one identified by reference numeral 104, receives transmissions from the mobile node, and communicates the transmissions to network elements within an associated regional access network 106. According to an embodiment, the regional access network 106 is a wired network (i.e., a physical line interconnects the various elements making up the regional access network) that is a generic packet based access network, such as an Ethernet network, an IP/MPLS network, or an ATM network. The transmission between the access points 104 and the mobile nodes 102 is compliant with Institute of Electrical and Electronics Engineers (IEEE) 802.16 standard signals, IEEE std. 802.16-2001, published 2001 and later versions (hereinafter IEEE 802.16 standard or IEEE 802.16e standard). A regional access network 106 interconnecting access points (such as 104) compliant with IEEE 802.16e standards is referred to as a WiMAX network.

At the periphery of the WiMAX regional access network 106 is a radio network services node 108. The radio network services node 108 provides routing and control between other WiMAX regional area networks, such as the WiMAX network identified by reference numeral 124. Each regional access network 106 and 124 includes a radio network services node (108, 110) that couples the regional access network 106 or 124 to a WiMAX core network 112, which interconnects all of the regional access networks 106 and 124. Although the WiMAX core network 112 is depicted in Figure 1 as interconnecting two WiMAX networks 106 and 124, the WiMAX core network 112 may, in principle, interconnect any number of regional access networks.

The WiMAX core network 112 may be an ordinary IP network, composed of commonplace IP network elements, such as optical networking elements permitting high speed data transfer. As such, the WiMAX core network 112 may interconnect directly with the Internet (not depicted in Figure 1).

At the periphery of the WiMAX core network 112 are one or more voice home agents 114 and 116. There exists a voice home agent 114 or 116 associated with each WiMAX regional access network 106 and 124. The structure of, and methods enacted by, a voice home agent 114 or 116 are discussed in detail below. Briefly, a voice home agent is a network element that permits VoIP integration between a WiMAX core network (such as core network 112) and the public switched telephone network (PSTN) 126.

Additionally, a voice home agent provides functionality that permits a mobile node (such as mobile node 102) to roam from one WiMAX regional access network (such as network 106) to another (such as 124).

Although Figure 1 depicts a single voice home agent 114 or 116 associated with each regional access network 106 or 124, more than one voice home agent may be associated with a given regional access network. Thus, although reference numerals 114 and 116 are presented herein as referring to a single voice home agent, each reference numeral 114 and 116 may be understood as referring to a group of voice home agents servicing their respective WiMAX regional access networks 106 and 124.

Each voice home agent 114 and 116 interfaces the WiMAX core network 112 to a local office 118 or 120 of the public switched telephone network 126. The public switched telephone network 126 uses an out-of-band signaling scheme known as Signaling System 7 (SS7), defined by the International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T). An out-of-band signaling scheme

employs a different physical path for call control than is used to carry the content of the call itself (e.g., the voice data). Therefore, as shown in Figure 1, a voice home agent serves as two separate interfaces: an interface for voice data, which is delivered as time domain multiplexed digital voice data, and an interface for SS7 control signals, which are delivered as SS7 packets.

A mobile node, such as the one identified by reference numeral 102, may be embodied as a telephone handset (in like fashion as a cellular telephone), may be embodied as a personal digital assistant, or may be embodied as another mobile computing device. Upon power-up, a mobile node makes an initial transmission to the nearest available access point. At the time of transmission, the access point assigns the mobile node a management channel, which identifies the mobile node to the access point. The access point and mobile node may communicate with another over a distance ranging from one to five or ten miles. Given the size of such an area, other mobile nodes may be located therein. Therefore, an access point may communicate with hundreds of mobile nodes. The use of management channels permits an access node to distinguish one access point from another.

Each access point in a WiMAX regional access network has an IP address that identifies it. However, this IP address is functional only within the regional access network (also referred to as a domain) in which the access point is situated. Thus, an access point may directly send data to another access point within the regional access network in which it is situated. To direct data to an access point in another domain, the radio network services node servicing the particular domain in which the access point is situated must be used as an intermediary.

As mentioned above, during power-up of the mobile node, an initial transmission is made to the base station for the sake of establishing a management channel and authenticating the user. Thereafter, the mobile node makes an initial communication with the voice home agent servicing the domain in which the mobile node is situated. This communication marks the beginning of a registration process, by which the mobile node informs the voice home agent of which domain the mobile node is in. In response, the voice home agent assigns the mobile node an IP address, known as a mobile IP (MIP) address. The voice home agent also records a care-of address for the mobile node. The MIP address for the mobile node does not change, even should the mobile node wander to a geographic region in which it communicates with another access point or with another

WiMAX regional access network altogether. The care-of address, on the other hand, identifies the domain with which the mobile node is communicating, and therefore changes when the mobile node roams from one regional access network to another.

5 A voice home agent may assign a mobile node more than one IP addresses. For example, a mobile node may have one IP address assigned to it for the carrying of voice data, and another IP address assigned to it for the carrying of signaling data. For the sake of simplicity, the disclosure proceeds from the assumption that each mobile node has a single IP address assigned to it during registration.

10 At the time of registration, the voice home agent updates a database that it maintains. The database may contain information concerning the features supported by the mobile node (call waiting, voicemail, etc.). The database is updated to associate a telephone number by which the mobile node is identified, the MIP address assigned to the mobile node, and the domain in which the mobile node is located (i.e., the care-of address of the mobile node).

15 A WiMAX regional access network 106 or 124 employs a technique known as tunneling. By virtue of this technique, movement of a mobile node within a geographic area served by a given WiMAX domain 106 or 124 is transparent to network elements or nodes outside of the domain. Thus, for example, a network node outside of WiMAX domain 106 cannot tell whether mobile node 102 is communicating with access point 104 or access point 122. A network element outside of the domain 106 need only know that the mobile node 102 is located in domain 106 to communicate with the mobile node 102. Therefore, whenever a mobile node (such as mobile node 102) moves from one domain to another, the mobile node re-registers with the voice home agent it previously registered with. In response, the voice home agent updates its database to associate a new care-of
20 address (i.e., network address of the domain with which the mobile node communicates) with the mobile node.
25

The preceding discussion focused on a network environment in which a voice home agent 114 or 116 operates. The following discussion briefly presents protocol layers making up a voice home agent 114 or 116.

30 Figure 2 depicts a protocol stack 200 executed by the voice home agent 114 or 116. As can be seen from Figure 2, the protocol stack 200 includes a Mobile IP (MIP) layer 202 that provides functionality complying with an industry-accepted MIP standard, such as the standard described in "IP Mobility Support," C. Perkins, ed., IETF RFC 2002,

Oct. 1996. The functionality provide by the MIP layer 202 is made available to the upper layers 204-210 of the stack 200.

The MIP layer provides the tunneling functionality mentioned above. Figure 3 depicts the MIP layer 202 receiving a packet 300 having an IP header 302. The IP header 302 contains the MIP address assigned to a particular mobile node in its 32-bit destination IP address field, and it therefore termed IP Header_{MIP}. In response to receiving such a packet 300, the MIP layer 202 appends the packet 300 to a second IP header 304. The second IP header uses the care-of address of the particular mobile node identified by the MIP address, and is therefore termed IP Header_{CareOf}. Thus, the WiMAX core network 112 observes the second IP header 304 and routes the packet 300 according to the second IP header 304, meaning that the packet 300 is routed to the appropriate domain 106 or 124. Prior to reception by the mobile node, the second IP header 304 is stripped away.

The effect of the tunneling technique described with reference to Figure 3 is that each mobile node receives IP packets containing the MIP address assigned to it during the registration process. Accordingly, each mobile node may roam—even roam between domains—while retaining the IP address assigned to it during the registration process.

Many layers of tunneling may be used in the network environment 100 depicted in Figure 1. For example, each WiMAX regional access network 106 and 124 may employ tunneling, so that elements laying outside the domain need only address IP packets to the proper domain in order for the packet to reach the desired mobile node.

Returning to Figure 2, it can be seen that the protocol stack 200 also includes a VoIP layer 204, which provides voice over IP functionality that may be compliant with an industry-accepted VoIP standard, such as Realtime Transport Protocol (RTP), which is defined by IETF RFC 1889 and/or Realtime Streaming Protocol (RTSP), which is defined by IETF RFC 2326. Briefly, the VoIP layer 204 receives VoIP packets and transforms those packets into time domain multiplexed digital voice data for the public switched telephone network (PSTN) 118 and 120, and *vice versa*. As discussed below, in the context of a discussion between a user of a mobile node and a user of the PSTN, the VoIP layer 204 converts time domain multiplexed digital data into VoIP packets. The VoIP packets contain the MIP address assigned to the particular mobile node. The VoIP packets are passed to the MIP layer 202, which appends the VoIP packets to an IP header containing the care-of address of the particular mobile node.

The protocol stack 200 also includes a session initiation protocol layer 206, which provides SIP functionality that may be compliant with an industry-accepted standard, such as IETF RFC 3261. Briefly, the SIP layer 206 provides application-layer control functionality for creating, modifying, and terminating communication sessions with one or more participants. For example, the SIP layer 206 contains the functionality to signal a mobile node that another party wishes to communicate with it.

The protocol stack 200 also includes a layer 210 that interfaces with the PSTN. The layer 210 includes a media gateway (MGW) that converts time-domain multiplexed voice data into IP packets. It also includes an SS7 interface that receives SS7 signals, decodes the signals, and passes the extracted information to the voice home agent control plane 208.

The voice home agent control plane 208 coordinates the actions of the other layers. It mediates communication between the major gateway and the VoIP layer 204, and also mediate communication between the SS7 interface and the SIP layer 206. For example, the voice home agent control plane 208 may receive a signal from the SS7 interface 210 indicating that a connection to a particular telephone number is desired. In response, the control plane 208 invokes the SIP plane 206 to send an SIP invite message to the mobile node corresponding to the telephone number. Similarly, the control plane 208 receives voice data in a particular time slot and forwards the data to the VoIP layer for conversion into VoIP packets, and for communication to particular mobile node (in this way, a voice path is maintained).

The preceding discussion briefly presented protocol layers 202-210 making up a voice home agent 114 or 116. A discussion relating to the operation of the voice home agent 114 or 116 with respect to call initiation and call execution follows. This discussion describes the operation of the voice home agent as a whole (as opposed to on a layer-by-layer basis), and provides an high-level integrated view of the operation of the voice home agent.

Figure 4 depicts the operation of a voice home 114 or 116 agent in initiating a telephone call to a mobile node. The process may be initiated by a user of the PSTN or by a user of a mobile node served by the voice home agent 114 or 116. If the process is initiated by a user of the PSTN, then the voice home agent 114 or 116 receives an SS7 signal indicating that a telephone call is desired with a mobile node identified by a particular telephone number, as shown in operation 400. The telephone number is

extracted from the SS7 signal (operation 400). The SS7 signal is converted into an invite message (operation 400), which is an SIP message indicating that a communication session is desired. Thus, at the completion of operation 400, the voice home agent 114 or 116 has constructed an invite message addressed to a particular telephone number.

5 On the other hand, the process may have been initiated by a mobile node served by the voice home agent 114 or 116. When a mobile node initiates the phone call, the mobile node sends an SIP invite message addressed to a chosen telephone number to the voice home agent 114 or 116. This SIP invite message is received by the voice home agent, as shown in operation 402.

10 Whether the SIP invite message is received (as is the case when a mobile node initiates the phone call) or is created by the voice home agent (as is the case when a user of the PSTN initiates the phone call), operation flow next proceeds to operation 404. In operation 404, the voice home agent queries a database to identify a MIP address and care-of address associated with the telephone number embedded in the invite message.

15 If the telephone number identified in operation 404 corresponds to the domain served by the voice home agent 114 or 116, then the voice home agent 114 or 116 sends the SIP invite message to the mobile node using the tunneling technique described with reference to Figure 3 (operation 406).

 If the telephone number identified in operation 404 corresponds to a domain not
20 served by the voice home agent 114 or 116, then the voice home agent 114 or 116 sends the SIP invite message to the voice home agent 114 or 116 serving the domain corresponding to the invited mobile (operation 408).

 If the telephone number indicates that the telephone number refers to a telephony device served by the PSTN, the invite message is converted to an SS7 signal to originate
25 the phone call on the PSTN (operation 410).

 After the invite message has been sent (by way of an SIP invite message or by way of an SS7 signal), the voice home agent 114 or 116 awaits a response to the invite message, as shown in operation 412. If the user of the invited telephony device wishes to answer the phone call, a response indicating such a desire is received by the voice home
30 agent 114 or 116 (operation 412). If the response originates from a mobile node, the response may reach the voice home agent 114 or 116 in the form of an SIP acknowledge (ack) message. On the other hand, if the response originates from a PSTN telephony

device, the response may come to the voice home agent 114 or 116 in the form of an SS7 signal that may be converted into an SIP ack message.

After the response is received, it is forwarded to the initiator (operation 414). If the initiator of the phone call is a mobile node, the forwarding operation involves sending the response to the mobile node, using MIP layer 202 to employ the tunneling technique described with reference to Figure 3. On the other hand, if the initiator of the phone call is a telephony device on the PSTN, then the response is converted into an SS7 signal, and is directed to the PSTN through the SS7 interface 210.

Finally, assuming the response received in operation 412 indicates that the user of the invited mobile node wishes to engage in a communication session (i.e., wishes to answer the call), a voice path between the inviting and invited devices is established (operation 416). Establishing the voice path may involve associating a particular time slot in the time-domain multiplexed voice data from the PSTN local office 118 or 120 with a particular MIP address (and vice versa). Additionally, it may involve associating an MIP address of a mobile node with a care-of address or an address of a voice home agent 114 or 116 servicing a particular mobile node.

After a VoIP session has been established (as shown in Figure 4), the parties may speak to one another. While the parties speak, the voice home agent 114 or 116 receives either VoIP packets or time-domain multiplexed voice data from the PSTN, as shown in operation 500 of Figure 5. If the voice home agent receives time-domain multiplexed voice data from the PSTN, such data is converted to VoIP packets, as discussed above.

Next, as shown in operation 502, the VoIP packets or voice data are sent along the voice pathway established in operation 416 of Figure 4. In the case of sending VoIP packets to a mobile node, this may mean sending received VoIP packets to a voice home agent 114 or 116 servicing the mobile node, or may mean directly sending received VoIP packets to a mobile node, using the tunneling technique described with reference to Figure 3. In the case of sending voice data to telephony device on the PSTN, operation 502 includes converting VoIP data to time-domain multiplexed digital voice data, and inserting such voice data into an appropriate time slot, so that the PSTN switching equipment routes the data to the appropriate location.

The preceding discussion related to the operation of the voice home agent during initiation and execution of a phone call. The following discussion presents, from a

system-level point of view, initiation and execution of a phone call in the context of the network environment 100 depicted in Figure 1.

In the context of a telephone call between a PSTN telephony device (initiator of call) and a mobile node (responder to call), the flow proceeds as follows. Initially, the voice home agent 114 or 116 receives an SS7 signal indicating that a communication session is desired with a mobile device corresponding to a given telephone number. The voice home agent extracts the telephone number, and creates an SIP invite message addressed to an MIP address of the invited mobile node. (If the invited mobile node is not available, the call may be re-routed to a voice mail service.)

By virtue of the tunneling capability of the voice home agent and the various WiMAX domains, the SIP invite message reaches the invited mobile node, addressed to the mobile node's MIP address. Within the SIP invite message, caller-ID information is embedded. Therefore, a message identifying the inviting telephony device may be displayed at the invited mobile node. Meanwhile, the voice home agent 114 or 116 sends an SS7 signal resulting in a ring-back tone to the inviting telephony device.

If the user of the mobile node accepts the call, then an SIP acknowledgement message is sent to the voice home agent 114 or 116. The voice home agent 114 or 116 translates the SIP acknowledgement message into an SS7 signal, and establishes a voice path. At this time, the users of the PSTN telephony device and the mobile node begin speaking.

In the context of a telephone call between two mobile nodes (in different domains), the flow proceeds as follows. Initially, the voice home agent receives an SIP invite message from the inviting mobile node. The SIP invite message is addressed to a telephone number corresponding with the desired mobile node. In response, the voice home agent forwards the SIP invite message to the voice home agent servicing the domain in which the invited mobile node is located. The latter voice home agent sends the response to the MIP address of the invited mobile node.

By virtue of the tunneling capability of the voice home agent and the various WiMAX domains, the SIP invite message reaches the invited mobile node, addressed to the mobile node's MIP address. Within the SIP invite message, caller-ID information is embedded. Therefore, a message identifying the inviting telephony device may be displayed at the invited mobile node. Further, the IP address of the inviting mobile node is contained in the SIP invite message.

If the user of the invited mobile node accepts the call, then an SIP acknowledgement message is sent to the voice home agent 114 or 116 servicing the domain in which the invited mobile node is located. In response, the voice home agent 114 or 116 forwards the SIP acknowledgement message to the voice home agent 114 or 116 servicing the domain in which the inviting mobile node is located. The latter voice home agent 114 or 116 forwards the SIP acknowledgement message to the inviting mobile node's MIP address. The SIP acknowledgement message contains the IP address of the invited node.

Voice communication may now occur in one of two manners. First, the mobile nodes may communicate with one another without the mediation of voice home agents. This is possible because, by virtue of the SIP invite and acknowledgement message, each mobile node is aware of the other's IP address. However, the connection between the two mobile nodes will be lost, should either of the mobile nodes roam to a different domain.

Secondly, the voice path may extend between both voice home agents. This scheme allows for either of the mobile nodes to roam from domain to domain.

Figure 6 depicts a hardware environment in which the voice home agent 114 or 116 may be embodied. The environment includes four blades 600, 602, 604, and 606. Each blade contains its own computing environment, including a processor, a memory, and an input/output module (control hub and I/O bus, for example) providing access to a network interface or to storage. Each blade 600-606 may communicate via a local area network, such as via an Ethernet hub. The blades 600-606 may be embodied as thin boards that may be mounted within a rack.

Each blade may be dedicated to executing various facets of the previously described control plane functions or data plane functions. For example, blade 602 may execute the functions relating to the MIP layer 202. This blade 602 also executes the routing functionality required when a VoIP packet is received, and needs to be routed to another voice home agent, or to a mobile node, as described above. The blade 602 includes a network interface to permit the software/firmware executed thereon to communicate with the WiMAX core network 112.

Blade 604 may execute the VoIP functionality described above with reference to the VoIP layer 204 discussed in Figure 2. The blade 604 includes a time domain multiplexing interface to permit the software/firmware executed thereon to interact with time domain multiplexed digital voice data from the PSTN.

Blade 606 may decode SS7 signals and send the extracted content to the SS7 application layer functionality residing on blade 600. The blade 606 includes an SS7 interface to permit the software/firmware executed thereon to interact with the SS7 packets from the PSTN local office.

5 Blade 600 may execute the voice home agent control plane functionality described above. For this purpose, the blade include a storage device (to maintain the database necessary to enact such functionality). The blade 600 also executes the SIP functionality and application layer functionality of the SS7 subsystem. In one embodiment, the blade 600 executes a billing routine. The billing routine may track, on a user-by-user or
10 account-by-account basis, the amount of time a given user is connected to the network, the amount of traffic consumed by the user, the type of service consumed (local call, long distance call, etc.) by the user, or the bandwidth consumed by the user. The tracked information may be stored in a database, and periodic bills may be generated therefrom.

Embodiments of the invention may be implemented in one or a combination of
15 hardware, firmware, and software. Embodiments of the invention may also be implemented as instructions stored on a machine-readable medium, which may be read and executed by at least one processor to perform the operations described herein. A machine-readable medium may include any mechanism for storing or transmitting information in a form readable by a machine (e.g., a computer). For example, a machine-
20 readable medium may include read-only memory (ROM), random-access memory (RAM), magnetic disc storage media, optical storage media, flash-memory devices, electrical, optical, acoustical or other form of propagated signals (e.g., carrier waves, infrared signals, digital signals, etc.), and others.

The Abstract is provided to comply with 37 C.F.R. Section 1.72(b) requiring an
25 abstract that will allow the reader to ascertain the nature and gist of the technical disclosure. It is submitted with the understanding that it will not be used to limit or interpret the scope or meaning of the claims.

In the foregoing detailed description, various features are occasionally grouped together in a single embodiment for the purpose of streamlining the disclosure. This
30 method of disclosure is not to be interpreted as reflecting an intention that the claimed embodiments of the subject matter require more features than are expressly recited in each claim. Rather, as the following claims reflect, inventive subject matter lies in less than all features of a single disclosed embodiment. Thus, the following claims are hereby

incorporated into the detailed description, with each claim standing on its own as a separate preferred embodiment.

Claims

The claimed invention is:

1. A device comprising:

encapsulation circuitry to receive an IP packet and prepend the packet with an IP header;

voice-over-IP (VoIP) circuitry to receive time-domain multiplexed voice data and
5 convert said data into VoIP packets;

signaling circuitry to decode Signaling System 7 (SS7) compliant signals; and
control circuitry to

receive decoded SS7 signals from the signaling circuitry and pass the
decoded SS7 signals to the encapsulation circuitry for transmission to a telephony
10 device; and

receive VoIP packets from the VoIP circuitry and pass the VoIP packets to
the encapsulation circuitry for transmission to the telephony device.

2. The device of claim 1, wherein the encapsulation circuitry, VoIP circuitry,
signaling circuitry, and control circuitry are embodied as a microprocessor in data
15 communication with a memory device.

3. The device of claim 1, wherein:

the encapsulation circuitry is embodied as a first blade, the VoIP circuitry is
embodied as a second blade, the signaling circuitry is embodied as a third blade, and the
control circuitry is embodied as a fourth blade; and

20 the first, second, third, and fourth blades are in data communication with one
another.

4. The device of claim 3, wherein the first, second, third, and fourth blades
communicate via an local area network.

5. The device of claim 3, wherein the fourth blade is further configured to maintain a database relating telephone numbers, mobile IP (MIP) addresses, and care-of addresses of mobile telephony devices.

6. The device of claim 3, wherein the first blade includes an interface to an IP
5 network.

7. The device of claim 3, wherein the second blade includes an interface to a network carrying time-domain multiplexed voice data.

8. The device of claim 3, wherein the third blade includes an interface to an SS7 network.

10 9. A method of conducting a voice-over-IP telephony session, comprising:
receiving a request, from a first telephony device, to invite a second telephony
device identified by a particular telephone number to participate in a voice-over-IP
telephony session;
relating the telephone number to an IP address associated with the telephone
15 number;
sending an invitation to the IP address;
receiving a response to the invitation; and
modifying and forwarding the response to the first telephony device, such that the
modified response contains a first IP header including a first IP address of the first
20 telephony device and a second IP header including a second IP address of the first
telephony device, wherein the first IP address indicates a point of connection of the first
telephony device to a network, and wherein the second IP address is generated as a result
of registration of the first telephony device with a voice home agent.

10. The method of claim 9, wherein relating the telephone number to an IP address
25 comprises accessing a database to determine an IP address of a voice home agent serving
the second telephony device.

11. The method of claim 9, wherein the invitation is compliant with a session initiation protocol (SIP).

12. The method of claim 9, further comprising:

receiving an IP packet containing data representing a voice; and

5 forwarding the IP packet to the first device using a third IP header including a third IP address, and a fourth IP header including a fourth IP address;

wherein the third IP address indicates a point of connection of the first telephony device to a network, and wherein the fourth IP address is generated as a result of registration of the first telephony device with a voice home agent.

10 13. The method of claim 9, further comprising:

receiving an IP packet containing data representing a voice; and

sending the IP packet to a voice home agent serving the second device.

14. The method of claim 9, further comprising:

15 determining that the first telephony device changed its point of connection to the network; and

redefining the first IP address to identify the changed point of connection.

15. The method of claim 9, wherein the first IP address identifies a radio network

20 services node coupled to an 802.16e-compliant network that includes an access point to which the first telephony device communicates.

16. A machine-accessible medium that provides instructions, which when accessed, cause the machine to perform operations comprising:

receiving a request, from a first telephony device, to invite a second telephony

25 device identified by a particular telephone number to participate in a voice-over-IP telephony session;

relating the telephone number to an IP address associated with the telephone number;

sending an invitation to the IP address;

30 receiving a response to the invitation; and

modifying and forwarding the response to the first telephony device, such that the modified response contains a first IP header including a first IP address of the first telephony device and a second IP header including a second IP address of the first telephony device, wherein the first IP address indicates a point of connection of the first
5 telephony device to a network, and wherein the second IP address is generated as a result of registration of the first telephony device with a voice home agent.

17. The medium of claim 16, wherein relating the telephone number to an IP address comprises accessing a database to determine an IP address of a voice home agent serving the second device.

10 18. The medium of claim 16, wherein the invitation is compliant with session initiation protocol (SIP).

19. The medium of claim 16, wherein the operations further comprise:
receiving an IP packet containing data representing a voice; and
15 forwarding the IP packet to the first device using a third IP header including a third IP address, and a fourth IP header including a fourth IP address;
wherein the third IP address indicates a point of connection of the first telephony device to a network, and wherein the fourth IP address is generated as a result of registration of the first telephony device with a voice home agent.

20 20. The medium of claim 16, wherein the operations further comprise:
receiving an IP packet containing data representing a voice; and
sending the IP packet to a voice home agent serving the second device.

21. The medium of claim 16, wherein the operations further comprise:
determining that the first telephony device changed its point of connection to the
25 network; and
redefining the first IP address to identify the changed point of connection.

22. The medium of claim 16, wherein the first IP address identifies a radio network services node coupled to an 802.16e-compliant network that includes an access point to which the first telephony device communicates.

23. A system comprising:

- 5 encapsulation circuitry to receive an IP packet and prepend the packet with an IP header;
- voice-over-IP (VoIP) circuitry to receive time-domain multiplexed voice data and convert said data into VoIP packets;
- billing circuitry that is configured to measure duration and type of use of said
- 10 system, and to relate such measurements to a user account;
- signaling circuitry to decode Signaling System 7 (SS7) compliant signals; and
- control circuitry to
- receive decoded SS7 signals from the signaling circuitry and pass the
- decoded SS7 signals to the encapsulation circuitry for transmission to a telephony
- 15 device; and
- receive VoIP packets from the VoIP circuitry and pass the VoIP packets to the encapsulation circuitry for transmission to the telephony device.

24. The system of claim 23, wherein the encapsulation circuitry, VoIP circuitry, signaling circuitry, billing circuitry, and control circuitry are embodied as a

20 microprocessor in data communication with a memory device.

25. The system of claim 23, wherein:

- the encapsulation circuitry is embodied as a first blade, the VoIP circuitry is embodied as a second blade, the signaling circuitry is embodied as a third blade, and the control circuitry and billing circuitry are embodied together as a fourth blade; and
- 25 the first, second, third, and fourth blades are in data communication with one another.

26. The system of claim 25, wherein the first, second, third, and fourth blades communicate via an local area network.

27. The system of claim 25, wherein the fourth blade is further configured to maintain a database relating telephone numbers, mobile IP (MIP) addresses, and care-of addresses of mobile telephony devices.

28. The system of claim 25, wherein the first blade includes an interface to an IP
5 network.

29. The system of claim 25, wherein the second blade includes an interface to a network carrying time-domain multiplexed voice data.

30. The system of claim 25, wherein the third blade includes an interface to an SS7 network.

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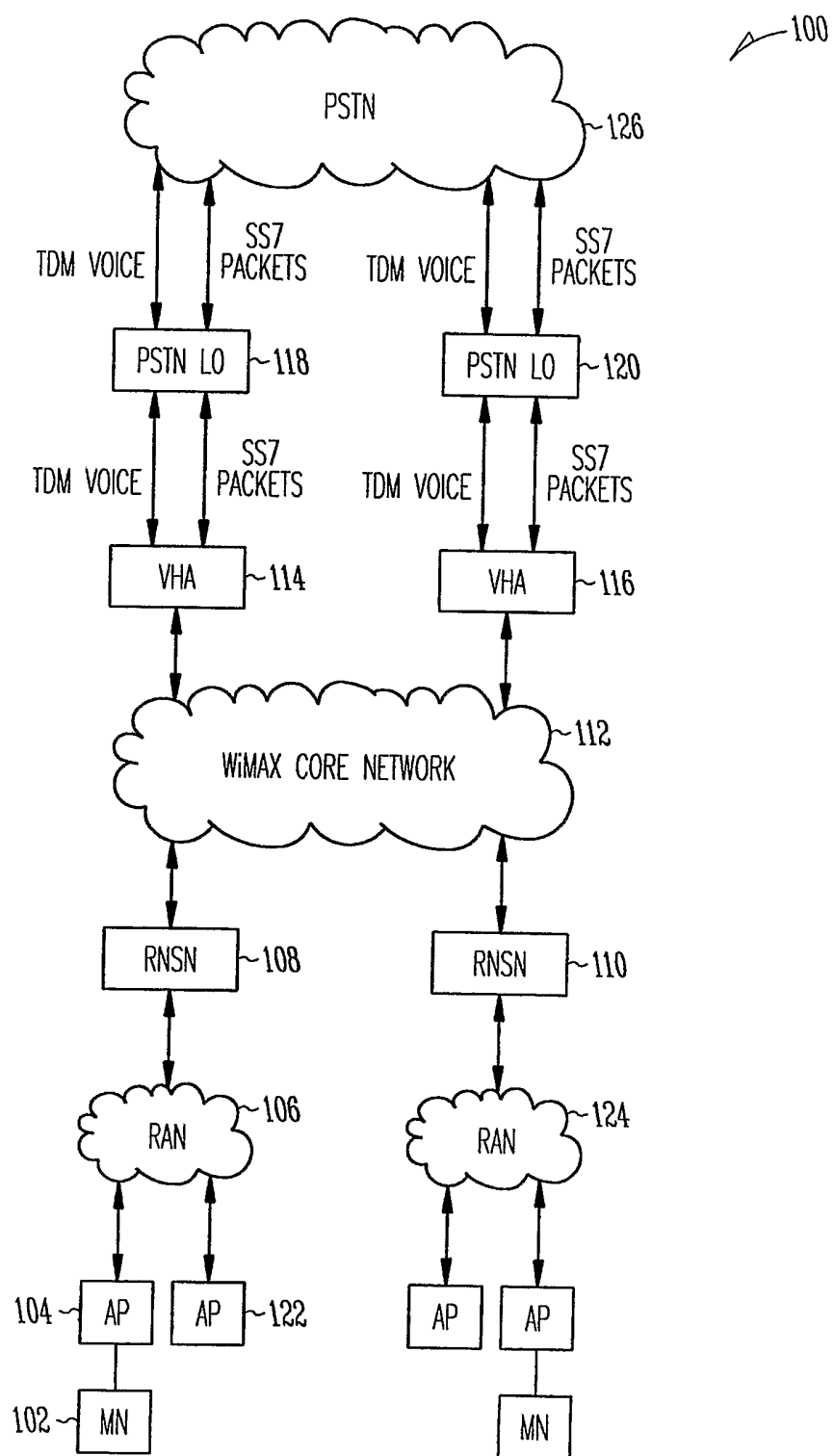


Fig. 1

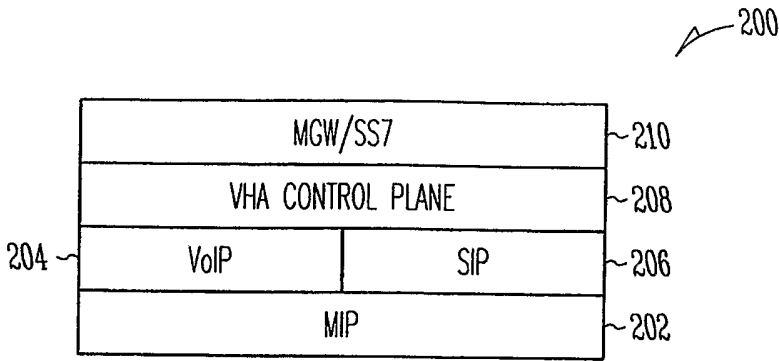


Fig. 2

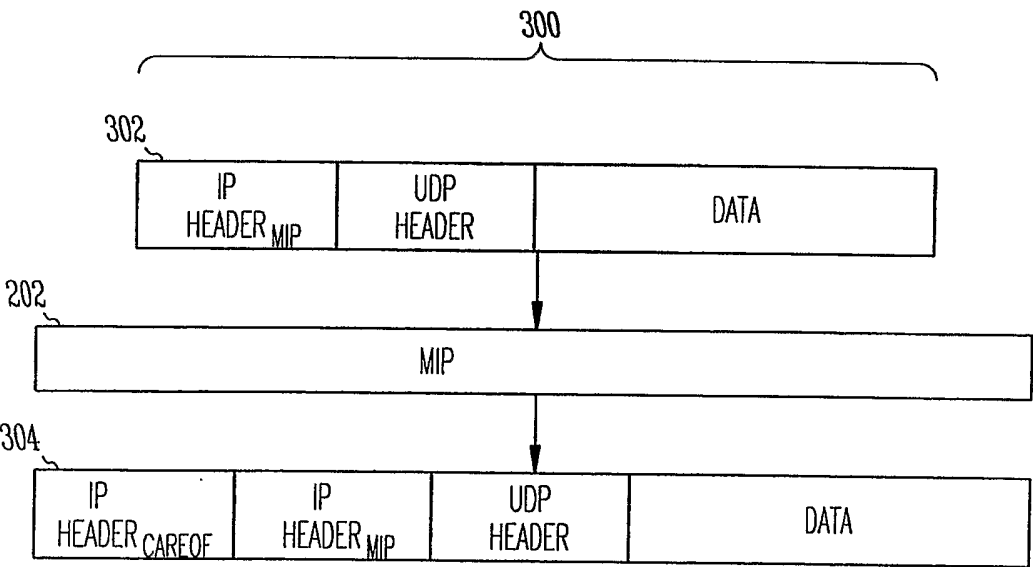
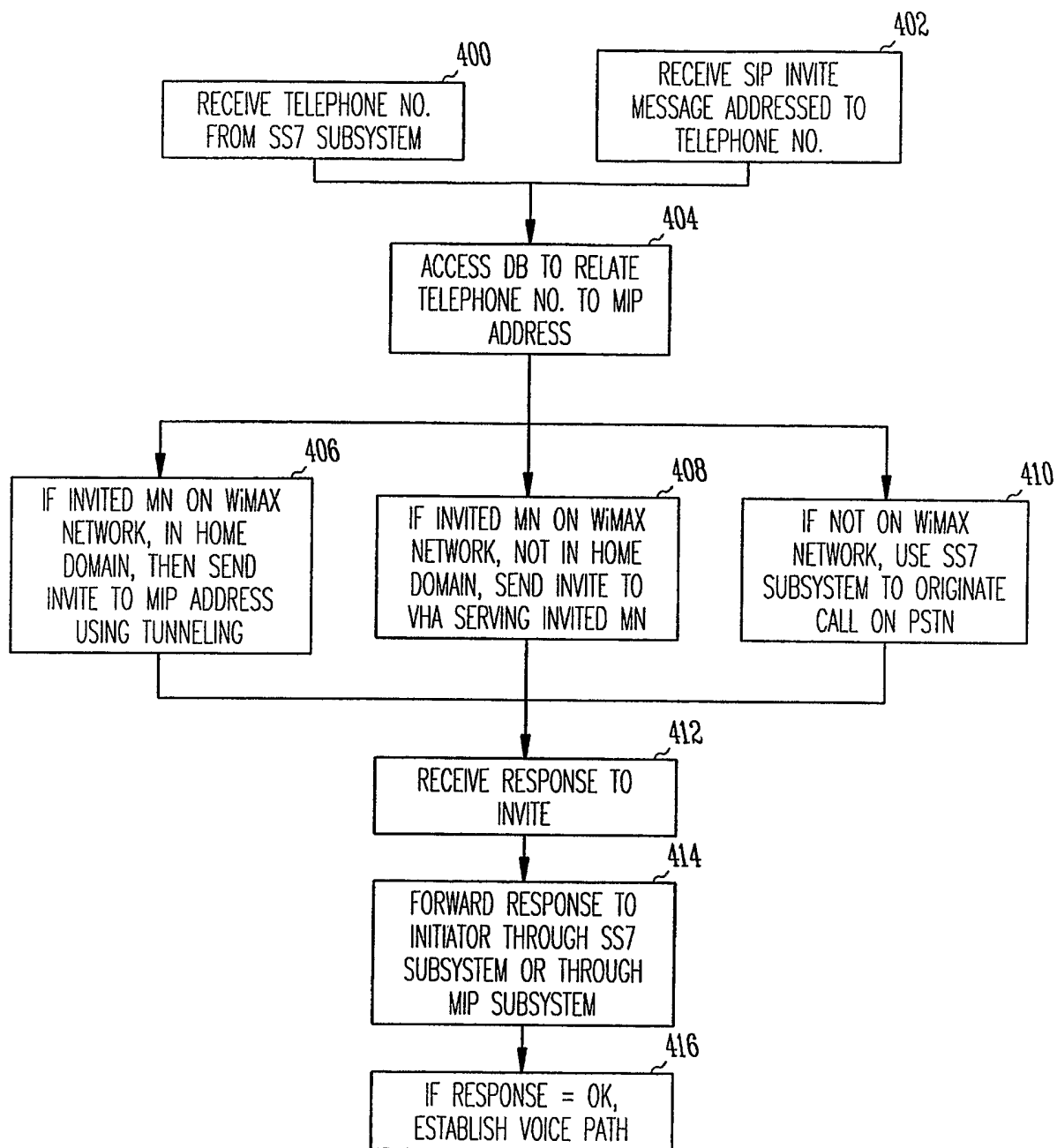
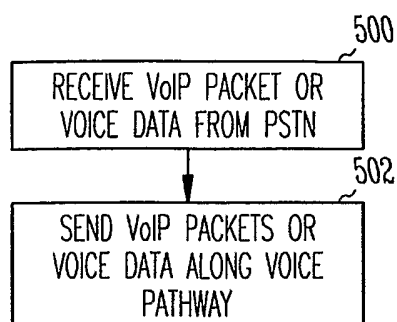
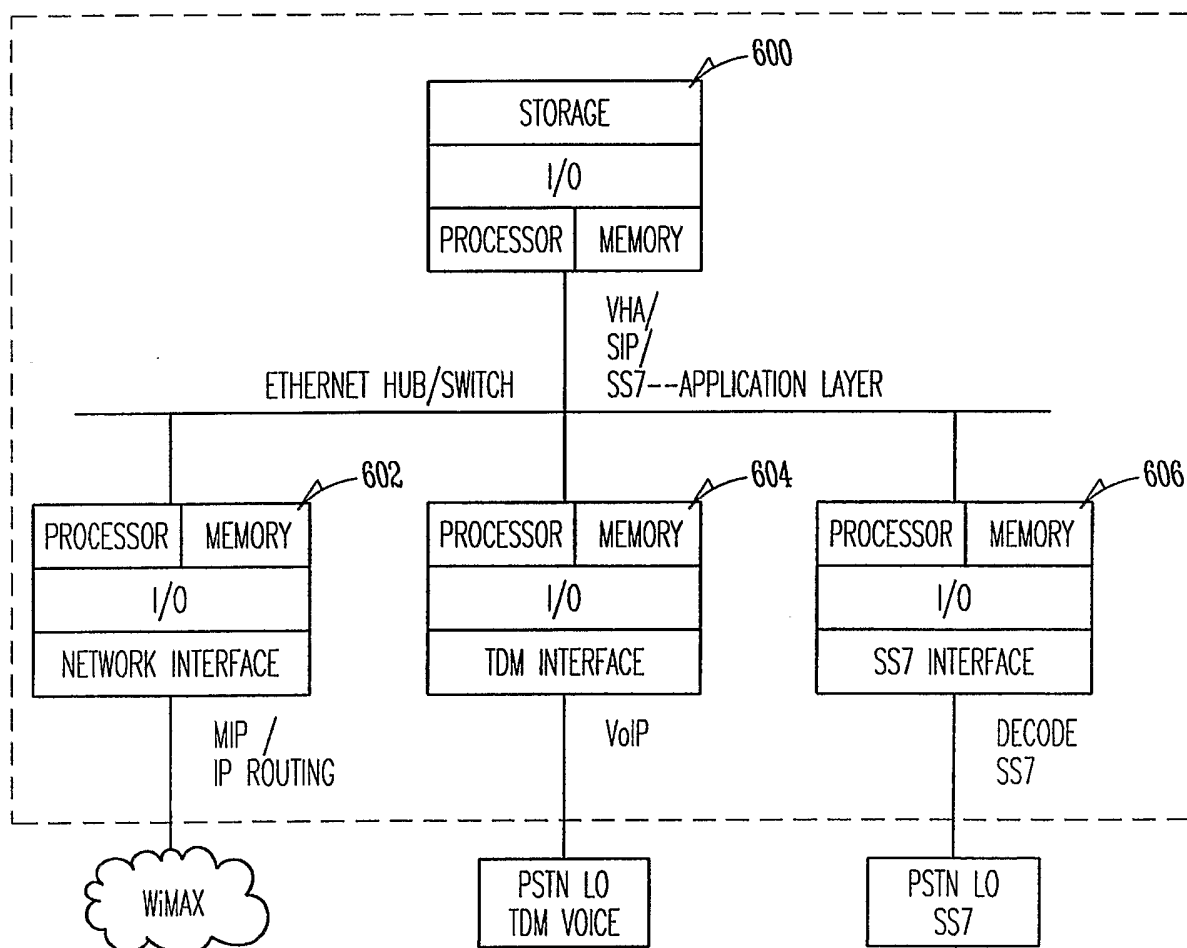


Fig. 3

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*Fig. 4*

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*Fig. 5**Fig. 6*

INTERNATIONAL SEARCH REPORT

International application No
PCT/US2005/047679

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04L29/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 00/79825 A (NOKIA NETWORKS OY; NOKIA INC.) 28 December 2000 (2000-12-28) page 1 - page 5 page 8 - page 25	1-30
X	NEWMAN P.: "IN SEARCH OF THE ALL-IP MOBILE NETWORK" IEEE COMMUNICATIONS MAGAZINE, IEEE SERVICE CENTER, NEW YORK, NY, US, vol. 42, no. 12, December 2004 (2004-12), pages S03-S08, XP001211443 ISSN: 0163-6804 page S03 - page S05	1-30

☒ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search

25 April 2006

Date of mailing of the international search report

04/05/2006

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INTERNATIONAL SEARCH REPORT

International application No
PCT/US2005/047679

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT		
Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	DUTTA A. ET AL.: "Realizing mobile wireless Internet telephony and streaming multimedia testbed" COMPUTER COMMUNICATIONS, ELSEVIER SCIENCE PUBLISHERS BV, AMSTERDAM, NL, vol. 27, no. 8, May 2004 (2004-05), pages 725-738, XP004501203 ISSN: 0140-3664 page 726 - page 733 -----	1-30
X	MORAND L ET AL: "Global mobility approach with mobile IP in all IP networks" ICC 2002. 2002 IEEE INTERNATIONAL CONFERENCE ON COMMUNICATIONS. CONFERENCE PROCEEDINGS. NEW YORK, NY, APRIL 28 - MAY 2, 2002, IEEE INTERNATIONAL CONFERENCE ON COMMUNICATIONS, NEW YORK, NY : IEEE, US, vol. VOL. 1 OF 5, 28 April 2002 (2002-04-28), pages 2075-2079, XP010589851 ISBN: 0-7803-7400-2 page 2075 - page 2077 -----	1-30
A	WO 01/67789 A (TELEFONAKTIEBOLAGET LM ERICSSON) 13 September 2001 (2001-09-13) page 16 - page 20 figure 8 -----	1-30

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US2005/047679

Box II Observations where certain claims were found unsearchable (Continuation of item 2 of first sheet)

This International Search Report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons:

1. ☐ Claims Nos.:
because they relate to subject matter not required to be searched by this Authority, namely:

2. ☐ Claims Nos.:
because they relate to parts of the International Application that do not comply with the prescribed requirements to such an extent that no meaningful International Search can be carried out, specifically:

3. ☐ Claims Nos.:
because they are dependent claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box III Observations where unity of invention is lacking (Continuation of item 3 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows:

see additional sheet

1. ☐ As all required additional search fees were timely paid by the applicant, this International Search Report covers all searchable claims.
2. ☒ As all searchable claims could be searched without effort justifying an additional fee, this Authority did not invite payment of any additional fee.
3. ☐ As only some of the required additional search fees were timely paid by the applicant, this International Search Report covers only those claims for which fees were paid, specifically claims Nos.:
4. ☐ No required additional search fees were timely paid by the applicant. Consequently, this International Search Report is restricted to the invention first mentioned in the claims; it is covered by claims Nos.:

Remark on Protest

- ☐ The additional search fees were accompanied by the applicant's protest.
- ☐ No protest accompanied the payment of additional search fees.

FURTHER INFORMATION CONTINUED FROM PCT/ISA/ 210

This International Searching Authority found multiple (groups of) inventions in this international application, as follows:

1. claims: 1-8 and 23-30

Claims 1-8 and 23-30 are directed to a tunneling VoIP / PSTN gateway. This is done by decoding SS7 signals, converting TDM voice into VoIP and further encapsulating everything into IP.

2. claims: 9-22

Claims 9-22 are directed to the conducting of a VoIP session using tunneling. This is done by receiving a request, from a first telephony device, to invite a second telephony device identified by a particular telephone number relating a telephone number to an IP address; sending an invitation to the IP address; receiving a response to the invitation; modifying and forwarding the response, such that the modified response contains a first IP header including a first IP address of the first telephony device and a second IP header including a second IP address of the first telephony device, wherein the first IP address indicates a point of connection of the first telephony device to a network, and wherein the second IP address is generated as a result of registration of the first telephony device with a voice home agent.

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No

PCT/US2005/047679

Patent document cited in search report		Publication date		Patent family member(s)	Publication date
WO 0079825	A	28-12-2000	AU	5096400 A	09-01-2001
WO 0167789	A	13-09-2001	AU	4832801 A	17-09-2001